

SYSTEM AND PROCESS FOR ROUTING TELEPHONY COMMUNICATIONS FROM
A CONVENTIONAL TELEPHONE SET THROUGH A DATA NETWORK

DESCRIPTION

5 TECHNICAL FIELD:

The present invention relates to telecommunications systems and in particular to the routing of telephone communications. The invention is especially applicable to a system and a process for routing telephony communications from a conventional ("PSTN") telephone set through a public data network, such as the Internet, to terminate, optionally, at another
10 conventional "PSTN" telephone; and to a subscriber interface unit for use therewith.

The invention is applicable whether access to the network is narrowband or broadband.

BACKGROUND ART

15 Most known systems for routing conventional telephony communications through public data networks generally fall into one of three classes. The first class of conventional routing system, usually used by large companies with private communications links, uses "re-diallers". Re-diallers re-route specific calls through a separate transmission path, e.g. a data network, using a real-time transport mechanism. This system requires the re-dialler to have
20 access to, and knowledge of, a private wide area data network, which is a significant disadvantage.

The second class of conventional routing system provides real-time voice communications from and to computer systems which have analog-to-digital conversion hardware and appropriate software; the user using a headset connected to his computer.
25 Recent modifications have allowed the use of a standard telephone set attached to the computer. These systems require the originator to establish a session with the data network, thus changing the user interface, and to have knowledge of the recipient's data address, e.g. IP (Internet Protocol) address. Both users require a computer with special equipment and each user must have special knowledge, limiting the usefulness of this class of product to the
30 general public. These are significant disadvantages.

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The third class of conventional routing system provides real-time voice communications from and to gateways located between a private branch exchange (PBX) and the public data network. This system requires a costly switching device between the PBX and the data network, thereby limiting its use to enterprises of a significant size. This precludes
5 use in a small business or home environment, which is a significant disadvantage.

United States patent number 6,205,135 (Venkata *et al.*) discloses a routing system using an alternate access platform located at a local exchange or, perhaps, at another location within the PSTN network that is further away from the subscriber. Limitations of such a system include the fact that the platform is expensive and hence not cost effective unless a
10 large number of subscribers access the platform to make Internet-based telephone calls. Moreover, if the company providing the Internet-based telephony services does not also own the local exchange, it will be charged a so-called collocation fee, leading to increased cost to the subscriber.

European patent application No. 1 061 728 A1 discloses a least-cost routing system
15 which is similar to those of the first class described above, but which uses a public data network, specifically the Internet, rather than a private network. Each subscriber has an IP router which establishes connections between the subscriber's telephone and either a telephone network provider or the Internet. When installing the IP router, the subscriber compiles his individual telephone directory with the telephone numbers and corresponding
20 IP addresses of all parties he may wish to call. When the subscriber dials a telephone number, the router uses it to access the individual telephone directory and determine whether there is a corresponding IP address. If there is none, the router uses least-cost-routing based upon the cost information to route the call over the conventional telephone network. If there is a corresponding IP address, however, the router determines whether the estimated cost
25 of the telephone connection via an Internet connection is more favourable than that of a normal telephone call and, if it is, routes the call as an IP call.

This is not entirely satisfactory, however, because storage of the telephone numbers and corresponding IP addresses at the subscriber access equipment adversely affects scaling of the system. Also, the use of previously-stored IP addresses may lead to connectivity
30 problems because IP addresses are not necessarily static. A further disadvantage is that

basing the route selection upon cost requires access to cost information which may entail subscriptions to several different service providers to obtain proprietary cost information.

DISCLOSURE OF INVENTION:

5 An object of the present invention is at least to ameliorate the disadvantages of such known systems, or at least provide an alternative.

According to a first aspect of the present invention, there is provided a method of routing telephone calls from a subscriber telephone in a telecommunications system comprising the Public Switched Telephone Network and a public data network, for example
10 the Internet, to a called party, the system including a local central office, a data network entrance node, a data network exit point, a subscriber interface unit located at the subscriber premises and connected between the subscriber telephone and the local central office, a called party interface unit connected to the data network exit point, a call server provisioned with data and software for routing of data calls between the data network entrance node and the
15 data network exit point, the call server data including destination numbers and corresponding network addresses of called party interface units that are capable of converting signals between PSTN and data formats, and a management server having software and data including rules for identifying classes of calls of a particular subscriber which should be routed via the data network, the subscriber interface unit having means for accessing said
20 rules, the method comprising the steps of:
at the subscriber interface unit,

- (i) detecting digits dialled by the subscriber to set up a call to said called party, said digits comprising at least the destination number of the called party;
- (ii) comparing at least some of the dialled digits with said rules to determine
25 whether or not, according to said rules, the call is included within a specified class of calls to be routed via the data network,
- (iii) if it is determined in step (ii) that the call not within the specified class, routing the call to a local central office for processing as a conventional PSTN telephone call;
- 30 (iv) if it is determined in step (ii) that the call is within a specified class and should be routed through the data network, connecting to the call server (38) and supplying

to the call server subscriber interface unit identification and the called party destination number;

at the call server,

5 (v) using the called party destination number, identifying a corresponding data network address for the called party interface unit, and setting up a data connection between the subscriber interface unit and the called party interface unit via the public data network according to the protocol of the public data network,

at each of the subscriber interface unit and called party interface unit,

10 (vi) on receipt of information that the connection has been established, converting subsequent signals from the subscriber and called party, respectively, from the PSTN format to a format compatible with the data network, and vice versa, as appropriate, until the call is terminated.

Preferred embodiments of the present invention provide individual subscribers with a subscriber interface unit capable of deciding whether to route a particular telephone call via the PSTN or a public data network on a per call basis using network selection rules, 15 conveniently supplied by the subscriber's service provider, that identify different classes of calls that should be routed via the data network. The decision process compares some of the dialled digits with the network selection rules to determine the class of call. Conveniently, the interface unit may compare only enough of the most significant digits of the called party 20 number to enable it to determine whether or not desired call falls within one of the classes defined by the rules. Hence, it is not necessary to use the entire destination number.

With such an arrangement, neither the subscriber nor the subscriber interface unit needs to have knowledge of the data network address of the called party. Moreover, the subscriber does not need to pre-program the telephone for data network calls by entering a 25 separate list of destination numbers to which calls may be made via the data network.

According to a second aspect of the present invention, a telecommunications system comprising the Public Switched Telephone Network and a public data network, for example the Internet, and means for routing calls between a subscriber telephone set and a called party, the system including a local central office, a data network entrance node, a data 30 network exit point, a subscriber interface unit located at the subscriber premises and connected between the subscriber telephone and the local central office, a called party

interface unit connected to the data network exit point, a call server provisioned with data and software for routing of data calls between the data network entrance node and the data network exit point, the call server data including destination numbers and corresponding network addresses of called party interface units that are capable of converting signals
5 between PSTN and data formats, and a management server having software and data including rules for identifying classes of calls of a particular subscriber which should be routed via the data network, the subscriber interface unit having means for accessing said rules, wherein:

the subscriber interface unit comprises,

10 means for (i) detecting digits dialled by the subscriber to set up a call to said called party said digits comprising at least the destination number of the called party and comparing at least some of the dialled digits with said rules to determine whether or not, according to said rules, the call is included within a specified class of calls to be routed via the data network;

(i)(a) if it is determined that the call is not within a specified class, routing the call to a local
15 central office for processing as a conventional PSTN telephone call; (i)(b) if it is determined that the call is within a specified class and should be routed through the data network, connecting to the call server and supplying subscriber interface unit identification and called party information to the call server;

the call server comprising means for identifying from the called party destination number the
20 network address for the called party interface unit and setting up a connection thereto; the subscriber interface unit and called party interface unit each further comprising means for converting subsequent signals from the subscriber and called party, respectively, from the PSTN format to a format compatible with the data network and routing the resulting data signals via the entrance and exit network nodes, and vice versa.

25 Preferably, where the subscriber interface unit is connected to the network node via a narrowband connection, the connection to the network node is initiated while the dialled digits are being collected and the routing decision made, so as to reduce waiting time for the subscriber.

Broadband embodiments of the invention may provide so-called "call waiting:", i.e.,
30 maintaining PSTN and VOIP calls simultaneously and allowing the subscriber to select one or the other without terminating either.

According to a third aspect of the present invention, there is provided a subscriber interface unit for use with the method of the first aspect, the subscriber interface unit having a first port for connection to a subscriber telephone set, a second port for connection to a subscriber line, means for detecting dialled digits received via the first port and using at least some of the dialled digits to access stored data including routing rules for different classes of calls to determine whether the call should be routed via the PSTN or via a data network and, in the former case, connecting the first port directly to the second port and, in the latter case, connecting the first port to the second port via conversion means for converting subscriber signals to data signal having a format suitable for routing through the data network and for converting data signals received from the data network into signals having a format compatible with the subscriber telephone set.

In embodiments of any of these three aspects of the invention, if the subscriber equipment supplies analog signals, the subscriber interface unit may perform analog-to-digital and digital-to-analog conversion of signals before and after the conversion to and from the data signals. Likewise, if the called party equipment is analog, the second interface unit may provide D-to-A and A-to-D conversion in addition to conversion to and from the data signal format.

Where the data network uses packet signals, the subscriber interface unit and the second interface unit may comprise means for converting from subscriber or called party signal format to packetized data signals, and vice versa.

The subscriber interface unit may include a conventional modem for converting the data signals into a format suitable for transmission to the local central office as a PSTN signal. Alternatively, where the subscriber has a high speed connection to a service provider, the conventional modem could be omitted and means provided for connecting the data conversion means, e.g. codec, to a high speed modem.

In preferred embodiments, the subscriber interface unit is installed at the subscriber's premises between the telephone set and the second interface unit is a PSTN gateway device connected to the called party equipment via the PSTN. Where the called party also is a subscriber to the routing service, however, the second interface unit could be similar to the subscriber interface unit.

The stored data accessed by the subscriber interface unit may be stored at the subscriber interface unit itself and periodically updated via the data network by a management application server unit at a remote location.

In preferred embodiments, the subscriber interface unit has a mechanical switch
5 connected between an input port and an output port for connection to the subscriber telephone set and the subscriber line, respectively, the switch being biased to connect the input port and output directly in the event of power failure to the subscriber interface unit.

Where the subscriber has a data connection to the local central office, the subscriber interface unit may have a port for connection to a high speed modem, in which case the
10 conventional internal modem can be omitted.

Preferably, when a call ends, the connection to the first network node is not taken down immediately in case the subscriber decides to make another call immediately. Rather, when a first call ends, the interface unit will monitor for the subscriber dialling new digits and, when such digits are detected, determine the routing for the new call, setting up a new
15 communication session, if appropriate, with the call server to set up a new connection to the new called party.

According to a preferred embodiment of the invention, there is provided a process for routing telephone calls including providing dial tone to a standard telephony device, collecting therefrom dialled digits representing a telephone number of a called party and using
20 at least some of the dialled digits to access stored data and determine whether or not to route the call over a public data network; connecting through the PSTN to a data network connection point (ISP), converting signals from the standard telephony device from analog to digital, converting the digital signals into packet signals with a format acceptable by the data network, routing the call via the data network to a data connection exit point convenient
25 for a called party's standard telephony device, receiving return digital signals from the called party standard telephony device via the data network, converting the return digital signals to analog and conveying the analog return signals to the subscriber telephony device.

Thus, the process is capable of routing telephony communications from a public switched telephone network (PSTN) over a public data network (Internet) and back to a
30 public switched telephone network (PSTN).

BRIEF DESCRIPTION OF THE DRAWINGS:

An embodiment of the invention will now be described by way of example only with reference to the accompanying drawings in which:

Figure 1 is a schematic diagram of a first embodiment of the present invention for
5 routing telephone calls within a public data network;

Figure 2 is a detail schematic view of a narrowband subscriber interface unit shown in Figure 1;

Figure 3 is a flowchart illustrating a first stage in the setting up of a call using the subscriber interface unit;

10 Figure 4 is a flowchart illustrating a second stage in the setting up of the call;

Figures 5A and 5B are flowchart sections illustrating the ending of the call in different ways;

Figure 6 illustrates a second embodiment of the invention for routing telephone calls through a public data network using a broadband connection;

15 Figure 7 illustrates a broadband subscriber interface unit of the system shown in Figure 6;

Figure 8 illustrates a first part of a sequence of setting up of a call using the broadband subscriber interface unit;

Figure 9 illustrates a second part of the sequence;

20 Figure 10 is a flowchart illustrating the ending of a call in different ways using the broadband subscriber interface unit;

Figure 11 illustrates the handling of an inbound call by the broadband subscriber interface unit; and

25 Figure 12 is a flowchart illustrating in more detail routing decision steps of Figures 3 and 8.

DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS

Figure 1 illustrates schematically deployment of a narrowband or "dial-up" embodiment of the invention in the public telecommunications system comprising a Public
30 Switched Telephone Network (PSTN) and a public data network, e.g., the Internet. A station apparatus (telephone set) 10 of a caller who is a service subscriber, i.e., who has paid

for the service provided according to the invention, is connected by way of a "router/dialler" or subscriber interface unit 12 (located at the subscriber premises) and subscriber loop 14 to a local central office of the Public Switched Telephone Network (PSTN) 16. The router/dialler unit 12 is installed at the subscriber's premises and is provided (leased or
5 purchased) to enable the subscriber to use the service.

The station apparatus 18 of a third party (the Called Party) is shown connected via a subscriber loop 20 to a remote or destination central office 22 in the PSTN 16. The local or originating central office 14 and the remote or destination central office 16 are interconnected via the PSTN, as represented by broken line 24.

10 Among other data, the router/dialler unit 12 stores the telephone number of a "HOME" Internet Service Provider (ISP) 26 accessible via the PSTN and data link 28. The HOME ISP 26, which is provisioned for Voice-over-Internet Protocol (VoIP), is shown connected via the Internet to an EXIT ISP 30, also provisioned for VoIP, which is coupled to a PSTN gateway 32 via a data link 34. The gateway 32 is coupled via link 36 to the
15 PSTN and therefore via the PSTN to the destination central office 22. (It should be noted that, in some situations, the HOME ISP 26 and the EXIT ISP 30 could be the same ISP.)

A call server 38 shown connected to the ISP 26 could be local to the ISP or at a remote location and accessed over the Internet or another data network. The call server 38 is provisioned with a dialled-digits store and routing data and software enabling it to route
20 calls to network nodes, including ISPs, in a known manner.

The data stored by router/dialler unit 12 is downloaded to router/dialler unit 12 during an initialization process carried out once it has been installed and connected, and then updated periodically, by a management application 40, namely software conveniently running on a personal computer acting as a server that is located on the premises of the ISP 26, or
25 at a remote location on the network, and able to communicate with the ISP 26 using the usual protocols. The router/dialler unit 12 is programmed with the telephone number of the ISP 26 and the name of the management application unit 40 (the name being used for routing purposes) and periodically calls the management application unit 40 for updates, identifying itself by means of its own unique identifier (i.e., serial number), like the unique identifier of
30 a mobile telephone. The management application unit 40 will download to the router/dialler unit 12 whatever information needs to be stored in the router/dialler 12 to enable it to

determine whether or not a particular call should be a VoIP call or a regular PSTN call. To this end, the management application 40 is provisioned with feature software enabling it to provide for Administration, Account Verification, Dialing Rule assignment/update and Router/dialler Setup.

5 The typical data that are downloaded by the management application unit 40 will include Network selection rules specifying that some calls are candidates for routing through the data network (VoIP calls) while others should be routed through the PSTN only. For example, international calls or other toll calls might be candidates for VoIP, whereas toll-free calls can be routed as PSTN-only calls. The actual selection would be at the discretion of the service provider, who might, for example, prefer to include toll-free calls as candidates for VoIP as well. Likewise, if local calls are free, as in most parts of North America, they too may be routed as PSTN-only calls.

The following table lists, as an example, a set of rules for making a routing decision between PSTN and VoIP networks. The table is generated by management application unit 40 on a per unit basis and downloaded to the SPIU 12, during either the initialization or a configuration update, where it is stored in memory.

The SPIU 12 collects incoming digits from the subscriber's telephone and compares them against network selection rules in its memory. If the destination number matches one of the rules, the call will be placed through the digital data network (VoIP); otherwise, if no match is found, the dialed number will be redialed to route the call via the PSTN.

The following table shows an example of the network selection rules of calls that are to be routed via the VoIP (data) network, all other calls, by virtue of their "absence", being routed via the PSTN by default:

25	All international calls 13-15 digit long	011-XXX-X-XXX-XXX[X][X]
	All North America calls with area codes from 205 to 613 inclusive	1-205-XXX-XXXX to 1-613-XXX-XXXX
	All North America calls with area codes from 904 to 925 inclusive	1-904-XXX-XXXX to 1-925-XXX-XXXX
30	All local calls started from 225 to 566	225-XXXX to 566-XXXX

According to this example of the rules, the following calls will be placed as follows:

Destination Number	Network
911	PSTN
411	PSTN
011-47-98-23-67-10	VoIP
011-44-20-7-230-1212	VoIP
1-108-345-5463	PSTN
1-503-751-4018	VoIP
1-715-645-1234	PSTN
1-800-123-4567	PSTN
1-900-234-8765	PSTN
1-914-342-5612	VoIP
324-3514	VoIP
735-1147	PSTN

15

It should be noted that, although the decision to route as a VoIP call is made by the subscriber interface unit 12, it does not store IP addresses of called parties. In making the routing decision, it is sufficient to compare the selected dialled digits with ranges of destination numbers, e.g. all 1 800 ### #### numbers, numbers between 1 205 ### #### and 1 613 ### ####, and so on, and route the call according to the rules.

When the subscriber begins to make a call, the router/dialler unit 12 will detect the dialled digits, consult the stored data (rules) and determine whether the call should be routed as a regular POTS call via the PSTN or as a Voice-Over-Internet-Protocol (VoIP) call, and route the call accordingly. This happens without the subscriber being involved in the decision.

If the router/dialler unit 12 determines that the call should be routed as a POTS call, it will simply route it via the central office 14 into the PSTN in the usual way (as represented by line 24).

If the router/dialler unit 12 determines that the call should be routed as a Voice-over-Internet-Protocol (VoIP) call, the router/dialler converts it into a data signal with the usual header etc. and routes it to the Internet in the usual way via HOME ISP 26, having first set up the connections by exchanging SIP messages with the call server 38. When setting up, the call server 38 detects the called party destination number in a "SIP Invite" message from the router/dialler 12, uses it to determine the closest gateway network node for the

destination, then establishes a connection to PSTN gateway node 32. Once the connection is established, the gateway node 32 supplies the gateway parameters, e.g. compression, CODEC type, and so on, to the router/dialler 12, via the call server 38. At the gateway node 32, the call is converted into a PSTN (analog voice) call again and routed via the PSTN to the destination central office 22, which completes the call in the usual way to the called party 18.

Referring now to Figure 2, the router/dialler 12 comprises a first port 42 to which the subscriber's telephone set 10 is connected, a mechanical relay 44, a microprocessor 46, a DSP/CODEC 48, a Hayes-compatible modem 50 (specifically a V90 56 kb/s modem), an PSTN port 52 which is connected to the subscriber line, a voltage regulator unit 54 connected to DC input socket 56 for connection to a separate power supply (not shown in Figure 2) and a Subscriber Line Interface Circuit (SLIC) 58. The voltage regulator 54 supplies the various components at the appropriate voltage. The SLIC 58 provides the subscriber telephone 10 with line voltage etc. which normally would be provided by the central office 14. The DSP/CODEC 48 provides digital-to-analog and analog-to-digital conversion, together with voice compression and decompression to ensure that the data signals are within the capability of the modem 50. The processor 46 controls the relay 44 and other components, as necessary, and controls the setting up and taking down of PPP communications with the HOME ISP 26 and SIP communications with the call server 38. In addition, the processor 46 converts the digitized voice signals to data signals, specifically VoIP signals, and vice versa.

Within the router/dialler 12, the subscriber's telephone set 10 is connected via input port 42 and mechanical relay 44 either to code 48 or directly to output port 52. If there is a power failure, or fault, the relay 44, by default, connects the telephone set 10 to the output port 52 via line 61 so the subscriber can make conventional calls, especially emergency calls, without involving the modem, etc. Moreover, prior to initialization of the recently-installed router/dialler unit 12, the processor 46 sets the relay 44 to connect the subscriber set directly to the output port 52.

Upon installation, and until the initialization procedure has been completed, all calls are placed to the PSTN. The initialization procedure is triggered when the subscriber lifts the receiver of the phone device 10 and dials a predetermined initialization sequence. This action

will initiate a VoIP connection setup with a call center representative. The call is then terminated at the call center and a customer service representative verifies all applicable customer profile information with the end user. Once that has been completed, the customer service representative initiates a procedure that provisions the management application server 5 40 with appropriate network selection rules and the call server 38 with authentication data (if required). The management application 40 then supplies the network selection rules, address of call server, configuration data, and so on, to the router/dialler 12 using a proprietary protocol. The received information is stored in the permanent unit memory and updated periodically by the management application 40, as necessary. Once the provisioning 10 of the router/dialler unit 12 is complete, it is ready for normal operation.

Assuming normal operating conditions with power supplied to the initialized router/dialler unit 12, the processor 46 will set the mechanical relay 44 to couple the telephone set 10 to the DSP/CODEC 48, i.e., the default presumes that any call will be VoIP. The processor 46 monitors the input port 42 so that, as soon as the subscriber picks up the 15 handset and starts to dial, the processor 46 can detect the "off-hook" condition.

Referring now to Figure 3, in which the activities by the subscriber are shown on the right and activities by the processor 46 are shown on the left, in step 3.01, the SLIC 58 detects the subscriber telephone set "going off-hook" in step 3.02 when the subscriber begins to make the call and reports the event to the processor 46 by way of the DSP/CODEC 48. 20 The setting up of a data call via an ISP takes a significant length of time, so the processor 46 does not wait for the subscriber to dial the called party's telephone before calling the HOME ISP 26. Rather, immediately the processor 46 receives the "off-hook" condition, in step 3.03 the processor 46 activates the modem 50 and, in step 3.04, detects via the modem 50 the usual dial tone supplied by the central office 14. In step 3.05, the processor 46 accesses its 25 stored data to obtain the telephone number of the HOME ISP 26 and dials it via modem 50.

It should be noted that the processor 46 has not yet connected the subscriber to the central office 14 so the subscriber does not hear the usual central office dial tone. Instead, in step 3.06, the processor 46 causes DSP/CODEC 48 to generate a "placebo" dial tone and 30 supply it via the SLIC 58 to the subscriber's telephone set 10. On hearing the placebo dial tone (in step 3.07), the subscriber dials the telephone number of the called party 18 (step

3.08). In step 3.09, the processor 46 collects the dialled digits, instructing the DSP/CODEC 48 to discontinue the placebo dial tone on receipt of the first digit, and in step 3.10 stores the digits temporarily in a buffer. Meanwhile, in step 3.11, the processor 46 is completing the usual "handshake" process with the HOME ISP 26, supplying login identification and
5 password as appropriate.

In decision step 3.12, the processor 46 accesses its stored data, specifically its "call routing rules", to determine whether the dialled digits represent a destination number to which the call should be routed as a PSTN-only call or as a VoIP call. If the rules specify that the call should be a PSTN-only call, in step 3.13 the processor 46 aborts the
10 establishment of the connection to the HOME ISP 26, or terminates it if completed, then, in step 3.14, retrieves the dialled digits from its buffer and relays them via modem 50 to the central office 14. In step 3.15, the processor 46 switches relay 44 to connect the subscriber's telephone set 10 to line 61 and port 52, thereby connecting the subscriber directly to the central office 14 to hear the usual central office ringing and other call progress tones (step
15 3.16). The call then continues as a regular PSTN-only call as indicated by step 3.17. If, in step 3.18, either party terminates the call, step 3.19 returns the processor 46 to scanning for originations (step 3.01).

Referring now to Figure 12, which illustrates the routing decision step 3.12 in more detail, step 12.01 sets a digit counter to zero and step 12.02 retrieves the corresponding digit
20 from the buffer (see step 3.10 of Figure 3). In step 12.03 the digit is compared with the appropriate digit from the network selection rules list. It should be noted that the numbers listed in the network selection rules are sorted in ascending numerical order. If step 12.05 determines that there is no match, step 12.06 causes the call to be placed via the PSTN.

If step 12.05 determines that there is match, step 12.08 determines whether or not
25 the current digit is the last digit (what constitutes "last" being determined by the network selection rules). If it is not, step 12.07 increments the digit counter and the processor repeats steps 12.02 to 12.08.

If and when step 12.08 determines that the current digit is the last digit, and step 12.09 determines that the appropriate dialling rule has been met, i.e., the destination number
30 has been collected and validated against the rules, decision step 3.12 (Figure 3) determines that the call should be routed as a VoIP call, and the router/dialler 12 performs the functions

illustrated in Figure 4. As shown in step 4.01, the processor 46 causes DSP/CODEC 48 to generate an "On Internet" tone and the SLIC 58 supplies it to the subscriber telephone set 10, to be heard in step 4.02. In step 4.03, the processor 46 accesses its stored data to determine the network name (address) of call server 38 and, using Session Initiation Protocol (SIP), initiates a call to it by sending it a "SIP Invite" message containing its unique identifier and the dialled digits. On receipt, the call server 38, in step 4.04, accesses its call server authentication data to confirm that the router/dialler 12 is used legitimately.

The call server authentication data is updated periodically by the management application 40 (Figure 1) with up to date authentication data. In step 4.05, the call server 38 uses its routing data to identify the appropriate gateway based upon this routing information. The routing information could be provided either by network administration (static routes) or automatic gateway location services (gatekeepers, ENUM and so on).

In step 4.06, the call server 38 establishes SIP-based communications with the PSTN gateway 32. In step 4.07, the PSTN gateway 32 completes the call to the called party 18 and notifies the call server 38 which supplies a valid termination message to the processor 46. RTP then is used to set up a bearer connection between the router/dialler 12 and the gateway 32 and ringing tone from the destination central office 22 is conveyed to the processor 46. In step 4.08, the processor 46 conveys the ringing tone via the DSP/CODEC 48 and SLIC 58 to the subscriber telephone set 10 (step 4.09).

When, in step 4.10, the called party answers, the call server 38 notifies the processor 46 which, in step 4.11, causes the DSP/CODEC 48 to discontinue the ringing tone. In step 4.12, the DSP/CODEC 48 begins A-D conversion and compression of the voice signal from the subscriber set 10 and the processor 46 converts the compressed digital signal to VoIP data for transmission by the modem 50 to the gateway 32. At the gateway 32, the VoIP data will be converted to PSTN format (step 4.13) and routed via the PSTN to the called party in the usual way.

Conversely, the PSTN gateway 32 digitizes the called party voice signals and converts them to VoIP sends them via the ISPs 30 and 26 to the router/dialler unit 12 where the processor 46 converts them from VoIP to digital and the DSP/CODEC 48 converts the digital signals to analog signal and routes the analog signal to the subscriber telephone set 10

via the relay 44. The call proceeds in this manner as indicated in step 4.14, and the processor 46 and gateway 32 monitor for one party or the other ending the call by "going on-hook".

Referring now to Figure 5A, step 5.01 constitutes one of the parties ending the call by "going on-hook" (this is the PSTN call on-hook). If it is the called party 18 who terminates the call and goes "on-hook" first, in step 5.02 the gateway 32 recognizes the "on-hook" condition and by a SIP message notifies the call server 32 which, on receipt of the notification in step 5.03 relays it to the processor 46 in router/dialler 12. On receipt of the notification in step 5.04 the processor 46 causes DSP/CODEC 48, in step 5.05, to provide a placebo dial tone to the subscriber telephone set 10. The SIP-based VoIP call has now been taken down.

The processor 46 does not immediately terminate the connection to the HOME ISP 26 but rather, in step 5.06, sets a timer and waits for a predetermined time while monitoring for (i) timer "time-out"; (ii) new dialled digits received, indicating that the subscriber has decided to make another call immediately, without hanging up the handset; and (iii) "off-hook" detected, to be explained later. If the timer times out, in step 5.07 the processor 46 will cause the DSP/CODEC 48 to discontinue the placebo dial tone and then, in step 5.08, will "tear down" the connection to the ISP; in step 5.09 wait for the subscriber to hang up and then proceed to step 3.01 (Figure 3) to wait for originations. Discontinuing the placebo dial tone will force the subscriber to hang up and go off-hook again should the subscriber wish to make another call.

If the subscriber, on hearing the placebo dial tone in step 5.10, decides to make another call and begins to dial a new telephone number before the timer times out, decision step 5.11 will take the path of "new digits dialled", in which case, in step 5.12, the processor 46 will cancel the timer, in step 5.13 cause the DSP/CODEC 48 to discontinue the placebo dial tone and then proceed to step 3.09 (Figure 3) to collect digits, then continue as before to set up the new call. In this case, the processor 46 will continue to maintain the connection to the ISP 26.

If the subscriber does not dial new digits but hangs up instead, from decision step 5.11 the processor 46 proceeds to cancel the timer in step 5.14 and then go to steps 5.07 to 5.09, causing the DSP/CODEC48 to discontinue the placebo dial tone, tearing down the

connection to the HOME ISP 26 and then proceeding to step 3.01 (Figure 3) to scan for originations.

Referring now to Figure 5B, if it is the caller/subscriber who hangs up first, in step 5.15 the processor 46 detects the "on-hook" condition and in step 5.16 notifies the call server 38 using SIP. On receipt of the notification, in step 5.17, the call server 39 will tear down the SIP connections to both the processor 46 and the gateway 32. The processor 46 does not terminate the connection to the HOME ISP 26 immediately, however, in case the subscriber decides to make another call immediately. Instead, the processor 46 goes to decision step 5.18, sets the timer and waits, while monitoring for the subscriber to go "off hook" again immediately and start dialling new digits or for the time-out to occur.

If no new "off hook" condition is detected before the timer times out, in step 5.19 the processor 46 terminates the connection to the HOME ISP 26, then returns to step 3.01 of Figure 3, i.e., resumes scanning for originations. On the other hand, if an "off-hook" condition is detected before the timer times out, decision step 5.18 takes the "off-hook detected" path, in which case, in step 5.20, the processor 46 will maintain the connection to the HOME ISP 26 and cancel the outstanding timer, then go to step 3.06 (Figure 3) to cause DSP/CODEC 48 to supply the placebo dial tone while the processor 46 collects dialled digits in step 3.09 and so on.

If desired for administration and billing purposes, the call server 38 will record when various events occur.

Operation of the router/dialler 12 to handle an incoming call is relatively straightforward. When the router/dialler 12 is in its "wait" condition, the processor 46 is monitoring not only the subscriber set 10 via the DSP/CODEC 48 and SLIC 50 for initiation of an outgoing call but also, via modem 50, the line 61 for indications of an incoming call. If the processor 46 detects an incoming call, it merely switches the relay 44 to connect line 61 directly to the subscriber's set 10. The processor 46 monitors line 61 to detect completion of the call at which point it causes relay 44 to return to its original state.

Referring again to Figure 1, if the call is destined for a called party 18B who has a similar broadband router/dialler unit 12A', the latter can communicate directly with the call server 38, i.e., without going through gateway 32, and itself convert from VOIP to analog. Thus, in this case, the codec units 48 in the two router/diallers 12A and 12A' will handle the

A-to-D conversion and compression, and vice versa, as appropriate. It would also be possible to complete the call to a called party with a SIP telephone or other compatible device elsewhere on the Internet, as indicated by optional ON NET EXTERNAL box 67.

A second embodiment of the invention, which used a broadband router/dialler, will now be described with reference to Figures 6 to 12. Components of the broadband system shown in Figure 6 which are the same as component in the embodiment of Figures 1 and 2 have the same reference numerals. Referring first to Figure 6, the subscriber telephone set 10 is connected to a broadband router/dialler 12A which differs slightly from the narrowband router/dialler 12 of Figures 1 and 2. As shown in Figure 7, the router/dialler 12A has three ports, a first port 42 for connection to the subscriber telephone set 10, a second port 52 for connection to the regular telephone line for transmission of PSTN-only analog telephone calls, and a third port 62 for connection via a high speed modem 63 (see Figure 6) to a broadband subscriber line.

The router/dialler 12A also differs from router/dialler 12 in that it has no Hayes-compatible modem; instead it has a Data Access Arrangement (DAA) unit 64 connected between the processor 46 and the output port 52 and a DTMF transceiver 65 connected between the processor 46 and the DAA 63. The DAA 64 interfaces with the central office 14 and provides various functions normally provided by the subscriber set 10, such as detecting voltage changes indicating that the central office 14 is "off-hook", detecting ringing tones, and so on. The DTMF transceiver 65 detects ringing tones from the central office and provides a means whereby the processor 46 can dial digits for transmission to the central office 14.

In Figure 6, port 52 is shown connected via subscriber loop 66 to the central office 14 of the PSTN 16 and port 62 is shown connected via high speed modem 63 to HOME ISP 26. It will be appreciated that, if the broadband connection is an Asynchronous Digital Subscriber Line (ADSL), both the high speed modem 63 and the regular PSTN port 52 would be connected via the same subscriber loop 66 to the central office 14 where a digital subscriber loop access multiplexer (DSLAM) would separate the PSTN-only calls and the data calls and route them to the PSTN 16 and the HOME ISP 26, respectively.

The high speed or data connection could, of course, be by telephone line, cable or a wireless/satellite connection. In all cases, however, it is presumed that the router/dialler 12A

does not need to set up a call to the HOME ISP 26 because it is in continuous communication with it.

Operation of the system of Figure 6 for different calls will now be described with reference first to Figure 8. In step 8.01, the processor 46 scans the subscriber telephone port 5 42 and line 61 for originations. If, in step 8.02, the subscriber picks up the handset to make a call, in step 8.03 the processor 46 causes the DSP/CODEC 48 to supply a placebo dial tone to the subscriber set 10. On hearing the placebo dial tone in step 8.04, the subscriber begins to dial the destination telephone number of the called party 18 (step 8.05). In step 8.06, the processor 46 collects the dialled digits and, in step 8.07, buffers them while determining the 10 routing for the call in step 8.08, using the rules previously stored in the router/dialler 12A.

As in the case of the narrowband router/dialler 12, the management application 40 downloads to the router/dialler 12A network selection rules and other data and updates it periodically. The detailed decision process carried out in step 8.08 is that illustrated in Figure 12 and previously described with reference to the narrowband router/dialler and decision step 15 3.12 (Figure 3), so it will not be described again here.

If the processor 46 determines that the call should be routed as a PSTN-only call, in step 8.09 the processor 46 opens a connection to the central office via DTMF transceiver 65 and DAA 64 and transmits to the central office 14 the dialled digits from its buffer. Once the call has been connected, in step 8.11 the processor 46 switches relay 44 to connect the 20 subscriber set 10 directly to the line 61 and thence to central office 14, whereupon, in step 8.12, the subscriber hears the ringing tone. In step 8.13, the call proceeds as a regular PSTN call until, in step 8.14, either party terminates whereupon, in step 8.15, the processor 46 returns to scanning for originations (step 8.01).

If, in step 8.08, the processor 46 determines that the call should be routed as a data 25 (VOIP) call, the processor 46 proceeds to step 9.01 and causes the DSP/CODEC 48 to supply to the subscriber set 10 an "On Internet" tone to be heard by the subscriber (step 9.02). In step 9.03, processor 46 accesses its stored data to determine the network name/address of the call server 38 and uses SIP protocol to initiate a call to the call server 38, supplying its own unique identifier and the destination number. In step 9.04, the call server 38 accesses its 30 authentication data to confirm that the router/dialler 12A is used legitimately. In step 9.05, the call server 38 uses its routing data to determine the network name/address of the

appropriate PSTN gateway 32 and in step 9.06 establish a SIP-based connection to the gateway 32 which completes the connection via the PSTN to the called party 18.

In step 9.07, the destination central office 22 supplies the usual valid termination message which the gateway 32 relays, using SIP, to the call server 38 and the processor 46.

5 On receipt of the valid termination message, in step 9.08 the processor 46 causes the DSP/CODEC 48 to supply to the subscriber 10 a ringing tone. In step 9.09, the call server 38 detects that the called party 18 has answered and relays the information to the processor 46 which, in step 9.10, instructs the DSP/CODEC 48 to discontinue the ringing tone and allows the call to proceed.

10 As shown in step 9.11, the DSP/CODEC 48 digitizes and compresses the analog signal from subscriber set 10 and the processor 46 converts the resulting digital signal to VOIP format and relays the VOIP signals via DAA 64, high speed modem 63, HOME ISP 26, PSTN gateway 32 and gateway 32 to the called party 18. As before, the gateway 32 converts the VOIP signal to analog form again before routing it to the called party 18.

15 Conversely, in step 9.12, the gateway 32 digitizes the analog signal from called party 18, converts it to VOIP format and routes it to the router/dialler 12A wherein the processor 46 converts it from VOIP to digital and DSP/CODEC 48 decompresses the digital signal, converts it to analog form, and conveys it to the subscriber 10.

As described with reference to the narrowband router/dialler 12, the calls from
20 broadband router/dialler 12A also could be made to a subscriber 18B or to a SIP phone 18C or other ON NET EXTERNAL device.

Figure 10 illustrates the sequence of operations when either of the parties ends the call, i.e., "goes on-hook" in step 10.01. If it is the called party 18 who terminates the call and goes "on-hook" first, the gateway 32 detects the event and issues a SIP message to notify the call
25 server 38. In step 10.02, the call server 38 recognizes the "on-hook" condition and, in a similar SIP message, notifies the processor 46 in router/dialler 12A. On receipt of the notification in step 10.03, the processor 46 proceeds to step 8.03 (Figure 8) to allow for further calls by the subscriber. At this point, the SIP-based connection is down.

If it is the caller/subscriber who hangs up first, in step 10.04 the processor 46 detects
30 the "on-hook" condition and in step 10.05 notifies the call server 38 using SIP. In step 10.06, the call server 38 receives the message to take down the SIP connection and proceeds to do

so according to SIP procedures. The processor 46 then goes to step 8.01 (Figure 8) to scan for originations.

As before, if desired for administration and billing purposes, the call server 38 will record when various events occur.

5 Operation of the broadband router/dialler 12A to handle an incoming call is relatively straightforward. As shown in Figure 11, when the router/dialler 12A is in its "scan" condition, step 11.01, the processor 46 is monitoring the subscriber set 10 for initiation of an outgoing call while the processor 46, via DAA 64 and DTMF transceiver 65, monitors the line 61 for indications of an incoming call. When the processor 46 and DAA 64 present on "on-hook"
10 condition, the central office transmits a "ringback" signal to the calling party and then the processor 46 in step 11.03 determines whether the subscriber set 10 is "on-hook". If it is, in step 11.04 the processor 46 connects the call to subscriber set 10 and in step 11.05 the ringing tone is heard by the calling party.

If step 11.03 determines that the subscriber set is "off-hook", in step 11.06 the
15 processor 46 determines whether or not there is a VOIP call in progress. If there is not, the processor 46 loops back to repeat steps 11.05 and 11.06 to periodically retests both conditions until it is positive or the inbound call is abandoned.

If step 11.06 determines that there is a VOIP call in progress, step 11.07 supplies alert tones for a predetermined period to the calling party. These alert tones, when heard by the
20 subscriber, inform the subscriber that there is an inbound call waiting. To accept the inbound call, the subscriber must terminate the VOIP call and then accept the inbound call by "going off-hook".

It will be appreciated that the broadband subscriber interface unit 12A shown in Figure 7 could be modified, specifically by programming of processor 46, to provide various special
25 telephony features, such as "call waiting" between both PSTN and VOIP calls. Because the processor 46 is connected to both the DAA 64 and the broadband connector 62, it is capable of maintaining both PSTN and VOIP calls and allowing the subscriber to switch from one to the other.

Although, in the above-described embodiments, the rules are stored in the subscriber
30 interface unit 12/12A and updated periodically by the management server 40, it is envisaged that the broadband version of the subscriber interface unit could access the management server

whenever it needed to access the rules, instead of storing them itself, and updating them periodically.

To summarize, in systems embodying the present invention, callers use a Voice-over-IP system to place outgoing calls over an ISP's network. Customers use their own telephone
5 instrument, plugged into a router/dialler, to access the ISP's network. The router/dialler connects to the ISP's network using a PPP connection when the customer places a call. The ISP operates a number of PSTN Gateways which bridge voice calls between the ISP's network and the PSTN. Calls are processed by a Call Server, which performs authentication, gateway selection and, optionally, call logging. A management application exists to enable
10 the service provider's support personnel to configure new router/diallers or update gateway lists.

Although, in the above-described embodiment, the public data network is the Internet, it will be appreciated that, with suitable modification, the invention could work with other data networks. It should also be appreciated that, although the above-described subscriber
15 interface unit is separate from the subscriber's telephone set, it could be incorporated into it. Hence, the invention comprehends a telephone set incorporating a subscriber interface unit as described herein for routing calls selectively via either the PSTN or a data network such as the Internet.

20 INDUSTRIAL APPLICABILITY

Advantageously, embodiments of the present invention provide selective routing of conventional POTS telephone communications via either the PSTN or a public data network without user intervention, facilitating optimization of route selection and transmission over the data network. This may reduce the cost of communication and avoid the need for a PBX-
25 to-data network switching device.

Embodiments of the invention advantageously enable standard telephone sets to communicate with other standard telephone sets by re-routing the communication through a public switched telephone system (PSTN) on to the public data system (Internet) and back onto the public switched telephone system (PSTN), thereby overcoming the requirement for
30 access to a private data network of previous systems.

An important advantage of embodiments of the invention is that no user intervention or knowledge of the called party's data network (IP) address is required when making the decision whether to route the call via the PSTN only or through interface with the data network. Hence, both broadband and dial-up embodiments of the invention advantageously
5 allow deployment of scalable and cost-effective VoIP services without requiring (i) high density of subscribers in each district; (ii) permits and license fees from telephone companies; and (iii) purchase of expensive Central Office equipment which is likely to be underutilized as a result of low-density of subscribers requesting VoIP. Moreover, embodiments support normal emergency calls (police, ambulance) using regular PSTN connections during power
10 outages. The subscriber interface unit is easy to install ("plug and play"), and does not require any manual configuration from the end user. The downloadable network selection rules allow for flexible subscription plans on a per customer basis, which allows the system to support only those VoIP calls that provide a competitive price advantage, for example "all long distance calls but not international calls". In addition, broadband embodiments of the
15 invention effectively provide the end-user with an additional phone line, which could be used by another telephone set in parallel with the PSTN line. Because neither the user nor the user's subscriber premises interface unit (SPIU) needs to have any knowledge of the recipient's IP address, connectivity problems related to changes in the network configuration are avoided.

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